CLAIMS

1. A decoder comprising:

an input for a number N of encoded input channels,

a matrix for transforming the input channels and implemented as a cascade of primitive matrix quantisers, the matrix providing the number N of matrix output channels,

a re-mapping arrangement arranged such that a number N of decoder output channels are ordered in response to channel ordering information derived from the input channels.

2. A decoder as claimed in claim 1, wherein the re-mapping arrangement is provided after the matrix.

3. A decoder as claimed in claim 1, wherein the re-mapping arrangement is provided before the matrix.

4. A decoder as claimed in claim 1, wherein the number N is six, and the encoded channels comprise the left-front, right-front, left-surround, right-surround, centre and low-frequency-effects channels of a six channel audio signal.

5. An encoder comprising:

a number N of input channels;

a reordering arrangement for permuting the channels;

a matrix for transforming the input channels and implemented as a cascade of primitive matrix quantisers, the matrix providing the number N of matrix output channels as a stream of encoded information formatted as a plurality of substreams, the data in the first substream comprising a strict subset of the outputs of the matrix and containing sufficient information for implementation of a downmix specification; and

means for encoding the selected permutation into the encoded output stream.

6. An encoder as claimed in claim 5, wherein the reordering arrangement is responsive to the downmix specification.

7. An encoder as claimed in claim 5 providing lossless compression, and further comprising at least one primitive matrix quantiser that modifies a channel that is not fed to the first substream, wherein the coefficients of that primitive matrix are chosen with the object of reducing the data rate of the encoded stream.

8. A computer memory product having stored thereon an encoded audio data signal, the encoded audio data signal comprising a number N of encoded audio channels which have been encoded by a matrix transformation, the encoded audio data further comprising channel ordering information, wherein the channel ordering information is provided for causing reordering of the channels at the input or output of a matrix transformation unit of a decoder for decoding the audio data.

9. An encoded audio data signal provided on a carrier, the encoded audio data signal comprising a number N of encoded audio channels which have been encoded by a matrix transformation, the encoded audio data further comprising channel ordering information, wherein the channel ordering information is provided for causing reordering of the channels at the input or output of a matrix transformation unit of a decoder for decoding the audio data.

10. A method of encoding a number N of input channels comprising:

permuting the channels;

applying a matrix transformation to the reordered input channels using a cascade of primitive matrix quantisers, the matrix providing the number N of matrix output channels as a stream of encoded information formatted as a plurality of substreams, the data in the first substream comprising a strict subset of the outputs of the matrix and containing sufficient information for implementation of a downmix specification; and

encoding the selected permutation into the encoded output stream,

wherein the permutation of the channels takes into account the downmix specification.

11. A method of decoding a number N of input channels, the method comprising:

applying an inverse matrix transformation to the input channels using a cascade of primitive matrix quantisers, the matrix providing the number N of matrix output channels;

obtaining channel ordering information from the encoded input channels; and

permuting the matrix output channels in dependence on the channel ordering information.

12. A decoder comprising:

an input for an encoded stream comprising a plurality of encoded channels;

at least a first primitive matrix quantiser that modifies a first channel;

a multiplier for multiplying the first channel by a gain coefficient; and

a combiner for combining the multiplied first channel signal with additional least significant bits recovered from the encoded stream,

wherein the gain coefficient is selectable by the decoder based on information in the encoded stream, and may take a number of values including one or more values which are not powers of two.

13. A decoder as claimed in claim 12, wherein the modified first channel is a linear combination of the channels input to the primitive matrix quantiser.

14. A decoder as claimed in claim 12, wherein the combiner comprises an adder.

15. An encoder for encoding a plurality of channels, the encoder comprising:

at least a first primitive matrix quantiser that modifies a first channel ;

a multiplier for multiplying the first channel by a gain coefficient within the first primitive matrix quantiser;

means for recovering any least significant bits which result from the multiplication which exceed a number of bits allocated to the channel; and

means for storing the recovered least significant bits, the multiplied modified first channel and a parameter representing the gain coefficient to enable

future decoding, wherein the gain coefficient may take a number of values including one or more values which are not powers of two.

16. A computer memory product having stored thereon an encoded audio data signal, the encoded audio data signal comprising a number N of encoded audio channels which have been encoded by a matrix transformation, the encoded audio data further comprising a parameter indicating a gain coefficient for at least one channel, and the encoded audio signal further comprising additional least significant bits for the at least one channel, wherein the gain coefficient is provided for causing multiplication of the channel within a matrix transformation unit of a decoder for decoding the audio data, and the additional least significant bits are provided for combination with the multiplied channel, and wherein the gain coefficient and may take a number of values including one or more values which are not powers of two.

17. A computer memory product as claimed in claim 16, wherein N=6.

18. An encoded audio data signal provided on a carrier, the encoded audio data signal comprising a number N of encoded audio channels which have been encoded by a matrix transformation, the encoded audio data further comprising a parameter indicating a gain coefficient for at least one channel, and the encoded audio signal further comprising additional least significant bits for the at least one channel, wherein the gain coefficient is provided for causing multiplication of the channel within a matrix transformation unit of a decoder for decoding the audio data, and the additional least significant bits are provided for combination with the multiplied channel, and wherein the gain coefficient may take a number of values including one or more values which are not powers of two.

19. A method of decoding an encoded stream comprising a plurality of encoded channels, the method comprising:

applying an inverse matrix transformation to the input channels using a cascade of primitive matrix quantisers, a first primitive matrix quantiser being used to modify a first channel;

within the first primitive matrix quantiser, multiplying the first channel by a gain coefficient derived from information in the encoded stream; and

combining the multiplied first channel signal with additional least significant bits recovered from the encoded stream, wherein the gain coefficient may take a number of values including one or more values which are not powers of two.

20. A method as claimed in claim 19, wherein the combining comprises adding.

21. A method of encoding a plurality of input channels, the method comprising:

applying a matrix transformation to the input channels using a cascade of primitive matrix quantisers, a first primitive matrix quantiser being used to modify a first channel;

within the first primitive matrix quantiser, multiplying the first channel by a gain coefficient;

recovering any additional least significant bits which result from the multiplication which exceed a number of bits allocated to the channel; and

storing the additional least significant bits, the multiplied modified first channel and a parameter indicating the gain coefficient to enable future decoding, wherein the gain coefficient may take a number of values including one or more values which are not powers of two.

22. An encoder apparatus for a lossless compression system furnishing an encoded compressed stream having at least two substreams, one of which provides a two-channel downmix signal and the other provides a multichannel digital signal, the encoder comprising:

an input for the multichannel digital signal;

an input for receiving a specification for the two-channel downmix signal; and

means for computing a first check value relating to a portion of downmix signal and inserting the first check value in a first substream and for computing a second check value relating to a portion of the multichannel signal and inserting the second check value in a second substream.

23. An encoder apparatus as claimed in claim 22 further comprising a downmix signal output enabling the downmix signal to be auditioned.

24. A decoder apparatus for decoding a losslessly compressed encoded signal, the decoder comprising:

an input for a multichannel encoded digital signal;

means for computing a first check value relating to a portion of the multichannel signal,

means for retrieving a second check value from the encoded stream relating to said portion of the multichannel signal;

a comparator arrangement for comparing the first and second check values; and

an output device for providing a flag indicating whether the check values agree.

25. An apparatus as claimed in any one of claims 22 to 24 comprising a digital signal processor for implementing each computation of a check value as a bitwise parity operation on the binary representations of the signal words in a segment of the signal, the binary words being rotated in dependence on the channel number before the parity is computed.

26. A player for a consumer disc comprising a decoder according to claim 24, wherein the output device comprises a visible indicator, and wherein the player further comprises a pulse-stretching circuit for providing a drive signal for the visible indicator, configured such that a lack of agreement between the first and the second check values for a time period shorter than a threshold level results in a visible indication for a time period longer than the threshold level.

27. An apparatus comprising an encoder as claimed in claim 22 and a prequantiser, the encoder providing as outputs an encoded compressed stream and a prequantised signal that can be auditioned, wherein the check value inserted in the compressed stream is computed from the prequantised signal.

28. A two-channel lossless decoder receiving as input an encoded stream , the decoder comprising:

a decoder core to which the stream is supplied to generate a two-channel decoded signal;

a shifter at the output of the decoder core for applying an arithmetic shift in response to information in the encoded stream; and

a signal limiting device at the output of the shifter, the shifter and the limiting device providing limitation of the wordwidth at the output of the decoder.

29. A decoder as claimed in claim 28, wherein the encoded input stream comprises at least two substreams, which together define a multi-channel encoded audio signal having more than two channels, and wherein one of the substreams is supplied to generate the two-channel decoded signal.

30. A decoder as claimed in claim 28 or 29 wherein the limiting device comprises a clipper which stores the output of the arithmetic shift using saturation arithmetic.

31. An encoder comprising:

an input for receiving a multi-channel signal having a first input channel or channels sampled at a first rate and a second input channel or channels sampled at a higher rate equal to twice the lower rate;

a device for increasing the sampling rate of the or each first input channel to the higher rate; and

a lossless encoder which receives the first and second channels at the higher rate, for generating an encoded stream.

32. An encoder as claimed in claim 31 further comprising means for inserting sample rate information into the encoded stream indicating, for each decoded channel, whether the corresponding input channel was sampled at the first or at the second rate.

33. An encoder as claimed in claim 32, wherein each device for increasing the sampling rate comprises half band filters providing a first plurality of samples corresponding to the input signal and a second plurality of interpolated samples.

34. An encoder as claimed in claim 33, wherein the means for inserting sampling rate information indicates which decoded samples correspond to the input signal.

35. An encoder as claimed in any one of claims 31 to 34 further comprising delay elements in the signal paths of the higher rate channels that substantially compensate the delays introduced by the sample rate increase of the lower rate channels.

36. A decoder comprising an input for an encoded stream produced by the encoder of claim 33 or 34 and furnishing a multichannel output, wherein the decoded samples of the lower-rate channels that do not correspond to original samples of the input channels are discarded, thereby furnishing an output identical to the multi-channel signal originally presented to the encoding apparatus.

37. A computer memory product having stored thereon an encoded audio data signal, the encoded audio data signal comprising a plurality of encoded audio channels representing channel signals sampled either at a first sampling rate or at a higher second sampling rate which is double the first sampling rate, wherein the encoded audio channels are all sampled at the second higher rate, and wherein additional information is provided for each channel indicating whether the channel signal was sampled at the first or the second rate.

38. A computer memory product as claimed in claim 37 wherein the additional information further indicates, for those encoded channels representing channel signals sampled at the first rate, which decoded samples correspond to the input signal.

39. An encoded audio data signal provided on a carrier, the encoded audio data signal comprising a plurality of encoded audio channels representing channel signals sampled either at a first sampling rate or at a higher second sampling rate which is double the first sampling rate, wherein the encoded audio channels are all sampled at the second higher rate, and wherein additional information is provided for each channel indicating whether the channel signal was sampled at the first or the second rate.

40. An encoded audio data signal provided on a carrier as claimed in claim 39, wherein the additional information further indicates, for those encoded channels

representing channel signals sampled at the first rate, which decoded samples correspond to the input signal.

41. An encoder comprising:

inputs for a multichannel digital signal and for a downmix specification, comprising; and

a matrix implemented as a cascade of primitive matrix quantisers and furnishing a stream of encoded information formatted as a plurality of substreams, the data in the first substream being taken from a strict subset of the outputs of the matrix and containing sufficient information for implementation of the downmix specification; wherein the matrix comprises:

a first primitive matrix having terms computed with respect to the specification for a first downmix channel, such that an output of the first primitive matrix is substantially the first downmix channel multiplied by a scaling factor; and

a second primitive matrix having terms computed such that a coefficient multiplying the output of the first primitive matrix is substantially zero.

42. An encoder as claimed in claim 41, wherein the first primitive matrix modifies an input channel having substantially the largest coefficient in the specification for the first downmix channel.

43. An encoder comprising:

inputs for a multichannel digital signal and for a downmix specification; and

a matrix implemented as a cascade of primitive matrix quantisers and furnishing a stream of encoded information formatted as a plurality of substreams, the data in the first substream being taken from a strict subset of the outputs of the matrix and containing sufficient information for implementation of the downmix specification, wherein the matrix defining the channels transmitted in the first substream in terms of the channels of the multichannel signal has rows that are substantially orthogonal to each other.

44. An encoder comprising:

inputs for a multichannel digital signal and for a downmix specification; and

a matrix implemented as a cascade of primitive matrix quantisers and furnishing a stream of encoded information formatted as a plurality of substreams, the data in the first substream being taken from a strict subset of the outputs of the matrix and containing sufficient information for implementation of the downmix specification, wherein the cross-correlation between the transmitted channels of the first substream is substantially zero.

45. An encoder as claimed in claim 44, wherein said cross-correlation is given a spectral weighting that emphasises the frequency bands that contribute substantially to the transmitted data rate.

46. An encoder as claimed in claim 45, wherein said cross-correlation is given a spectral weighting substantially as defined by the digital filter $(1 - z^{-1})^n$, where $n = 1, 2$ or 3.

47. In a lossless encoder comprising two or more input channels and transmitting output signals that are linear combinations of the input channels, a method of determining the output signals for transmission, comprising the steps of:

(i) establishing a first list of input channels, and a second list defining output signals determined for transmission;

(ii) deleting one channel from the first list and choosing a linear combination of the remaining channels in the first list and of the output signals determined for transmission, the linear combination being chosen substantially to minimise an entropy measure for a difference signal equal to the difference between said one channel and the linear combination;

(iii) applying a scaling factor to the difference signal to obtain a scaled difference signal;

(iv) allocating the scaled difference signal as an output signal for transmission, and including a definition of that output signal in the second list; and

(v) repeating steps (ii) to (iv) for a further input channel or channels.

48. A method as claimed in claim 47, wherein the definition of the output signal comprises the output signal itself.

49. A method as claimed in claim 47, wherein the definition of the output signal comprises matrix specifications enabling the output signal to be calculated.

50. A method as claimed in claim 47, wherein the first list is initially equal to the entire set of input channels, and the second list is initially empty.

51. A method as claimed in claim 47, wherein the second list initially contains signals that allow a downmix to be recovered, and the first list contains a subset of the set of input channels.

52. A method as claimed in claim 47, in which the step of deleting one channel comprises selecting a channel for deletion from the first list for which the absolute magnitudes of the coefficients in the linear combination do not exceed a predetermined limit.

53. A method as claimed in claim 47, in which the step of deleting one channel comprises selecting a channel having the smallest energy.

54. A method as claimed in claim 47, in which said scaling factor is +1 or −1.

55. A method as claimed in claim 47, in which said scaling factor is +½ or −½.

56. A method as claimed in claim 47, in which said entropy measure is energy.

57. A method as claimed in claim 47, in which said entropy measure is energy calculated with a spectral weighting that emphasises the frequency bands that contribute substantially to the data rate of a compressed output stream of the encoder.

58. A method as claimed in claim 47, in which said entropy measure is energy calculated with a spectral weighting substantially as defined by the digital filter $(1 - z^{-1})^n$, where $n = 1, 2$ or 3.

59. A method as claimed in claim 47, in which calculation of the scaled difference signal is performed by a primitive matrix quantiser.

60. A method as claimed in claim 47, in which the scaling factor has a magnitude of less than unity, and in which additional least significant bits recovered following the scaling operation are provided as an output signal for transmission.

61. A method of encoding two or more channels to produce output signals for transmission that are linear combinations of the input channels, the method comprising:

determining the output signals for transmission using the method of any one of claims 47 to 60; and

calculating the determined output signals.

62. A method of encoding a multichannel signal containing input channels and a downmix specification, comprising the steps of:

choosing a first input channel having substantially the largest coefficient in the specification of a downmix channel; and

scaling down the first input channel using a primitive matrix quantiser in which the coefficient of the first channel is less than unity and the coefficients of the other input channels are substantially zero.

63. A method as claimed in claim 62, in which additional least significant bits recovered following the scaling operation are provided as an output signal for transmission.